IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

Applicant: Oh et al
Art Unit: 2654
Serial No.: 09/483,569
Examiner: Michael N. Opsasnick
Filed: January 14, 2000
Docket: TI-23373
For: SIMPLIFIED NOISE SUPPRESSION CIRCUIT

Appeal Brief under 37 C.F.R. §41.37

Board of Patent Appeals and Interferences
United States Patent and Trademark Office
P.O. Box 1450
Alexandria, VA 22313-1450

Dear Sir:

This is Appellant's Appeal Brief filed pursuant to 37 C.F.R. §41.37 and the contemporaneously filed Notice of Appeal.

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Real Party in Interest

The real party in interest in this application is Texas Instruments Incorporated, a corporation of Delaware with its principal place of business in Dallas, Texas. An assignment to Texas Instruments Incorporated is recorded at reel 010513 and frames 0488 to 0490.

Related Appeals and Interferences

There are no appeals of interferences related to this appeal in this application.

Status of the Claims

Claims 1 to 3 and 9 to 11 are finally rejected. Claims 4 to 8 and 12 to 22 are canceled. No claims are allowed.

Status of Amendments Filed After Final Rejection

No amendments to the claims were proposed following the FINAL REJECTION of July 14, 2006.

Summary of Claimed Subject Matter

This invention is a method and apparatus for reducing noise in a sampled acoustic signal. A sampler (104, page 7, lines 8 and 9) obtains discrete samples of an acoustic signal. An analog to digital converter (106, page 7, lines 10 to 11) forms a stream of sampled acoustic signals. The invention selects a fixed number of samples (200, page 7, lines 27 to 28; 300, page 8, lines 31 to 35). This fixed number of samples is preferably 32 samples. The invention multiplies these samples by a windowing function (202, page 7, lines 28 to 31; 302, page 9, lines 7 to 13). This windowing function is preferably a Hanning window function. A fast Fourier transform of the windowed samples yields transformed windowed signals (204, page 7, line 31 to 33; 304,

page 9, lines 15 and 15). The invention selects half of the transformed windowed signals (page 7, line 34 to page 8, line 1; 306, page 9, lines 15 and 16). The invention calculates a power estimate of the transformed windowed signals (page 8, lines 1 and 2; 308, page 9, lines 16 to 23) and a smoothed power estimate by smoothing the power estimate over time (page 8, lines 2 to 5; 310, page 9, lines 23 to 31). The invention calculates a noise estimate (page 8, line 7; 312, page 9, line 31 to page 10, line 23). Then the invention calculates a gain function from the noise estimate and the smoothed power estimate (page 8, lines 7 to 9; 314, page 10, line 24 to page 11, line 10). The invention calculates a transformed speech signal by multiplying the gain function with the transformed windowed signal (page 8, lines 9 to 11; 316, page 11, lines 11 to 14). An inverse fast Fourier transform (208, page 8, line 15 to 16; 318, page 11, line 15) of the transformed speech signal yields a sampled speech signal. The invention adds the sampled speech signal to a portion of the speech signal of a previous frame (320, page 11, lines 16 and 17).

Grounds for Rejection to be Reviewed on Appeal

Claims 1 to 3 and 9 to 11 were rejected under 35 U.S.C. 103(a) as made obvious by the combination of Bloebaum et al U.S. Patent No. 6,070.137 and Oppenheim (Discrete Time Signal Processing, pp 57, 59, 60, 542, 543, 548).

Arguments

Claims 1 to 3 and 9 to 11 were rejected under 35 U.S.C. 103(a) as made obvious by the combination of Bloebaum et al. U.S. Patent 6,070,137 and Oppenheim (Discrete Time Signal Processing).

Claims 1 and 9 recite subject matter not made obvious by the combination of Bloebaum et al and Oppenheim. Claim 1 recites "calculating a smoothed power estimate by smoothing the power estimate over time." Claim 9 recites the noise suppression circuit operates to "calculate a smoothed power estimate by smoothing the power estimate over time." The FINAL REJECTION demonstrates that Bloebaum et al fails to make this limitation obvious. In particular, the FINAL REJECTION shows that Bloebaum et al teaches smoothing over time of a different signal than that claimed in claims 1 and 9.

Claim 1 recites calculation of "a gain function from the noise estimate and the smoothed power estimate." Claim 9 recites the noise suppression circuit operates to "calculate a gain function from the noise estimate and the smoothed power estimate." The FINAL REJECTION states at page 3, lines 7 and 8 that Bloebaum et al teaches:

> "calculates a gain function from the signal and noise power estimates (enhancement filter, col. 6, lines 8-10), and"

This portion of the FINAL REJECTION refers to Figure 4 of Bloebaum et al. Figure 4 illustrates transform and filter computation block 56 receiving the power spectral density (PSD) estimate represented by $|S(e^{j?})|^2$ from block 44 and the noise vector N from noise model adaptation block 46 and producing enhancement filter $|H(e^{j?})|$. In order for the Examiner's statement at page 3, lines 7 and 8 of the FINAL REJECTION to be true, one input to transform and filter computation block 56 must

correspond to the claimed noise estimate and the other input must correspond to the claimed smoothed power estimate. Bloebaum et al states at column 5, lines 58 and 59:

> "The forward transform G converts the noise vector N into the noise PSD estimate $|N(e^{j?})|^2$."

Thus this input to transform and filter computation block 56 must correspond to the claimed noise estimate. Accordingly, the other input to transform and filter computation block 56 $|S\hat{}(e^{j?})|^2$ must correspond to the claimed smoothed power estimate for the Examiner's statement to be correct. However, Bloebaum et al fails to teach that this input is smoothed over time as required by the language of claims 1 and 9. Bloebaum et al states at column 5, lines 60 to 62 referring to variance reduction block 58:

> "The Variance Reduction block receives as input $|S(e^{j?})|^2$ and applies a smoothing function in the frequency domain to generate an output $|S\hat{}(e^{j?})|^2$."

Thus Bloebaum et al clearly teaches $|S\hat{}(e^{j?})|^2$ is smoothed in the frequency domain and not smoothed over time as recited in claims 1 and 9. The description of the claimed time smoothing in this application states at page 9, lines 23 to 26:

> "After that is accomplished, the power estimate is smoothed over a time index (as opposed to a spectral smoothing as is used in the spectral substraction method) in step 310."

This "spectral smoothing" is the frequency smoothing disclosed in Bloebaum et al. Thus the text of the application states that the claimed time smoothing is different from the frequency smoothing taught in Bloebaum et al. The Applicant respectfully submits

that disclosure of smoothing in the frequency domain fails to make obvious the smoothing over time of claims 1 and 9.

The FINAL REJECTION states at page 3, lines 2 to 6 that Bloebaum et al teaches:

> "calculating a smoothed power estimate over time by smoothing the power estimate using the recited (i.e., first-order AR smoothing) equation (Fig. 5, element 64 with 'smoothed version of S' in col. 8, lines 6-8; *cf.* first order AR smoothing, col. 5, lines 38-44), wherein noting that S is signal power with signal present and noise power when signal absent, thus also calculating a noise estimate,"

The Applicants submit that the signal N supplied to transform and filter computation block 56 from noise model adaptation block 46 is only a noise estimate and includes no signal. Bloebaum et al states at column 5, lines 21 to 45:

> "An important aspect of integrating noise suppression into the MBE speech encoder 20 is the computation of a model of the background noise. The noise model in FIG. 3 is represented as a vector N output from a noise model adaptation block 46. This invention is not restricted to any particular method of modeling background noise, and several possible methods are discussed herein. The noise model is stored by the noise model adaptation block 46 and is updated when the vadFlag is set to zero, indicating that there is an absence of speech. The adaptation process involves smoothing of the model parameters in order to reduce the variance of the noise estimate. This may be done using either a moving average (MA), autoregressive (AR), or a combination ARMA process. AR smoothing is the preferred technique, since it provides good smoothing for a low ordered filter. This reduces the memory storage requirements for the noise suppression algorithm. The noise model adaptation with first order AR smoothing is given by the following equation:
>
> $$N^{(i)} = aN^{(i-1)} + (1-a)S,$$
>
> where a may be in the range 0.1=a=1, but is further constrained to the range 0.8=a=0.95 in the preferred

> embodiment of the invention. The vector S is an input to block 46 from a Transform and Filter Computation block 56."

The text of Bloebaum et al makes clear that the vector N is a noise model "output from a noise model adaptation block 46." The first order AR smoothing of the equation is used in adapting the noise model. This portion of Bloebaum et al teaches that the noise model "is updated when the vadFlag is set to zero, indicating that there is an absence of speech." Accordingly, the AR smoothing equation is employed only in the absence of signal in S and is employed only to update a "noise model is stored by the noise model adaptation block 46." This portion of Bloebaum et al clearly teaches smoothing of the vector N from noise model adaptation block 46 as a function of the prior noise vector N and the vector S in the absence of signal. Thus this is not smoothing the power estimate as claimed. Claims 1 and 9 recite such a noise estimate as a different signal employed in the calculation of the gain function. This equation of Bloebaum et al fails to make obvious calculating "a smoothed power estimate by smoothing the power estimate over time" as recited in claims 1 and 9.

The FINAL REJECTION cites variance reduction 64 described in Bloebaum et al at column 8, lines 6 to 8 and illustrated in Figure 5 as teaching the recited smoothing over time with reference to Bloebaum et al at column 5, lines 38 to 44. Bloebaum et al at column 5, lines 38 to 44 teaches smoothing over time of the noise vector N produced by noise model adaptation block 46. This smoothing over time is not applicable to variance reduction 64 of Figure 5. Bloebaum et al states at column 8, lines 1 to 10:

> "This alternate version is denoted by block 62 and is shown in FIG. 5. The principal novelty of the block 62 versus the

> block 56 is that the enhancement filter is computed in the domain of the noise model and then transformed to the sampled frequency domain. In FIG. 5, the signal model vector S is input to the Variance Reduction block 64, which outputs a smoothed version of S denoted S^. This vector S and the noise model vector N are input to the Enhancement Filter Computation block 66."

This teaching of Bloebaum et al fails to state that variance reduction block 64 smoothes over time as required by the language of claims 1 and 9. Because Figure 5 is taught as an alternative to Figure 4, one skilled in the art would believe that variance reduction block 64 operates similarly to analogous variance reduction block 58 of Figure 4. As quoted above, Bloebaum et al states at column 5, lines 60 to 62 variance reduction block 58 smoothes in the frequency domain. Accordingly, one skilled in the art would believe that variance reduction block 64 also smoothes in the frequency domain. Thus claims 1 and 9 are not made obvious by the combination of Bloebaum et al and Oppenheim.

The FINAL REJECTION at page 3, line 17 to page 4, line 6 states that the disclosure in Oppenheim that convolution in the frequency domain corresponds to multiplication in the time domain means that the smoothing in frequency domain taught in Bloebaum et al makes obvious the smoothing in the time domain recited in claims 1 and 9. The FINAL REJECTION states that Bloebaum et al teaches at column 6, lines 1 to 6 that the smoothing in the frequency domain can be performed by linear or circular convolution in the frequency domain. Employing the teaching of Oppenheim with this teaching of Bloebaum et al means that the convolution computed smoothing in the frequency domain taught in Bloebaum et al is the equivalent to multiplication in the time domain. Note that claims 1 and 9 recite smoothing in the time domain and do not recite multiplication in the time domain. This equivalence to a different function taught in Oppenheim fails to

make obvious the smoothing in the time domain recited in claims 1 and 9. Oppenheim teaches that convolution in the frequency domain is equivalent to multiplication in the time domain. The Examiner attempts to use Oppenheim to teach that smoothing in the frequency domain taught in Bloebaum et al is equivalent to the smoothing in the time domain recited in claims 1 and 9. This is not the teaching of Oppenheim and is incorrect. Note further that Oppenheim does not teach smoothing in the time domain and likewise fails to teach what would be its equivalent function in the frequency domain. Thus the equivalence taught in Oppenheim when applied to the disclosure of Bloebaum et al fails to make obvious the time smoothing of the power estimate recited in claims 1 and 9. Accordingly, claims 1 and 9 are allowable over the combination of Bloebaum et al and Oppenheim.

In summary, Bloebaum et al teaches a calculation of a gain or filter function in transform and filter computation block 56 similar to the recitations of claims 1 and 9. In Bloebaum et al, one input $|N(e^{j?})|^2$ is related to the noise estimate and the other input $|S\hat{}(e^{j?})|^2$ is related to the power estimate. Claims 1 and 9 recite smoothing over time of the power estimate related input. Bloebaum et al teaches smoothing over time of noise estimate related term. This time smoothing is of the other input than that recited in claims 1 and 9. In addition, Bloebaum et al teaches smoothing in the frequency domain of the power estimate related input. This differs from the time domain smoothing language of claims 1 and 9. Oppenheim fails to teach the equivalence of smoothing in the frequency domain of Bloebaum et al with smoothing the time domain recited in claims 1 and 9. Thus while this application states at page 9, lines 23 to 26 that frequency domain smoothing and time domain smoothing are not the same, no teaching of the combination of references teaches that

these are equivalent. Accordingly, the combination of Bloebaum et al and Oppenheim fails to make obvious claims 1 and 9.

Claims 2, 3, 10 and 11 are allowable by dependency upon respective allowable base claims 1 and 9.

If the Examiner has any questions or other correspondence regarding this application, Applicants request that the Examiner contact Applicants' attorney at the below listed telephone number and address to facilitate prosecution.

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CLAIMS APPENDIX

1. A method for reducing noise in a sampled acoustic signal, comprising:

receiving a stream of sampled acoustic signals;

digitizing each sampled acoustic signal thereby forming digital samples;

selecting a fixed number of digital samples;

multiplying the digital samples by a windowing function;

computing the fast Fourier transform of the selected windowed digital samples to yield transformed windowed signals;

selecting half of the transformed windowed signals;

calculating a power estimate of the transformed windowed signals;

calculating a smoothed power estimate by smoothing the power estimate over time using the equation:

$$P^{t}(i) = (1-a)\ P^{t-1}(i) + a\ P(i)$$

where: $P^{t}(i)$ is the smoothed power estimate for a current time sample to be calculated for the i-th FFT point; $P^{t-1}(i)$ is the smoothed power estimate for an immediately prior time sample for the i-th FFT point; P(i) is the calculated power estimate of the transformed windowed signals for the i-th FFT point; and a is an experimentally chosen predetermined value called the smoothing factor;

calculating a noise estimate;

calculating a gain function from the noise estimate and the smoothed power estimate;

calculating a transformed speech signal by multiplying the gain function with the transformed windowed signal;

calculating an inversed fast Fourier transform of the transformed speech signal to yield a sampled speech signal; and

adding the sampled speech signal to a portion of the speech signal of a previous frame.

2. The method of Claim 1, wherein the fixed number of samples is thirty-two.

3. The method of Claim 1, wherein the windowing function is a hanning window function.

9. A system for reducing noise in an acoustical signal comprising:

a sampler for obtaining discrete samples of the acoustical signal;

an analog to digital converter coupled to the sampler an operable to convert the analog discrete samples into a digitized sample;

a noise suppression circuit coupled to the analog to digital converter and operable to:

receive the digitized samples;

select a fixed number of digitized samples;

multiply the digitized samples by a windowing function;

compute the fast Fourier transform of the windowed digitized samples to yield transformed windowed signals;

select half of the transformed windowed signals;

calculate a power estimate of the transformed windowed signals;

calculate a smoothed power estimate by smoothing the power estimate over time using the equation:

$$P^{t}(i) = (1-a)\ P^{t-1}(i) + a\ P(i)$$

where: $P^{t}(i)$ is the smoothed power estimate for a current time sample to be calculated for the i-th FFT point; $P^{t-1}(i)$ is the smoothed power estimate for an immediately prior time sample for the i-th FFT point; P(i) is the calculated power estimate of the transformed windowed signals for the i-th FFT point; and a is an experimentally chosen predetermined value called the smoothing factor;

calculate a noise estimate;

calculate a gain function from the noise estimate and the smoothed power estimate;

calculate a transformed speech signal by multiplying the gain function with the transformed windowed signal;

calculate an inversed fast Fourier transform of the transformed speech signal to yield a sampled speech signal; and

add the sampled speech signal to a portion of the speech signal of a previous frame.

10. The system of Claim 9, wherein the fixed number of samples is thirty-two.

11. The system of Claim 9, wherein the windowing function is a hanning window function.

Evidence Appendix

None

Related Proceedings Appendix

None